SYNTHESOURCE

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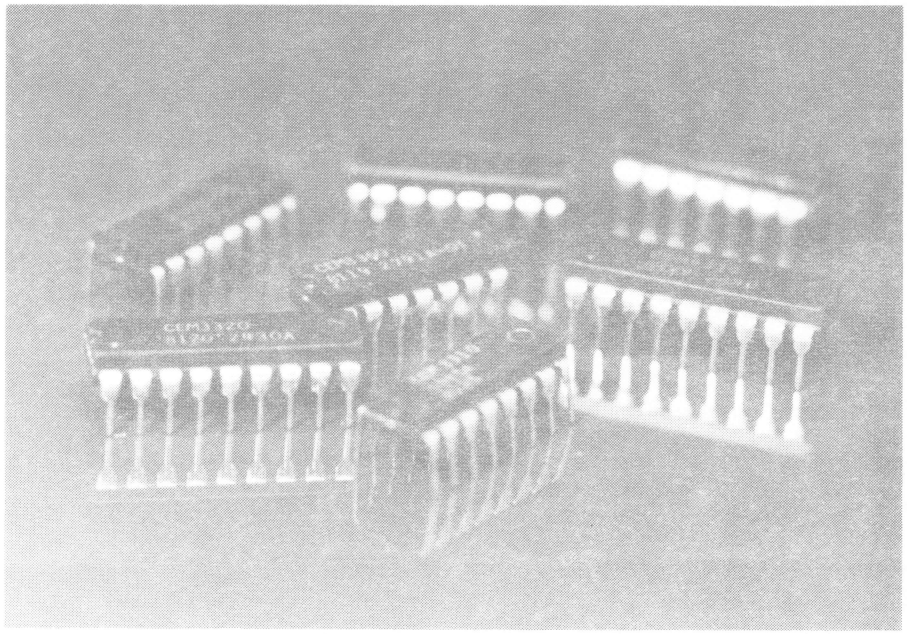
CEM 3350 Debuts New Filter Sounds

PRODUCT HIGHLIGHT

CEM 3350 VCF Configurations Explored in Depth

We at Curtis Electromusic have always been big fans of filters. Not only do they add great variety of tone color to those raw VCO waveforms, but they also allow for easy dynamic timbre changes within notes, so important for adding richness to music generation. However, it is also our conviction that the standard four pole low pass filter, which has served the industry almost exclusively since the pioneering days, is not the only game in town. Of course, there are many good reasons why the old 4PLP remains so popular, but there certainly must be other types of filters and their responses which are just as musically valid and useful.

Hence we designed the CEM 3350 dual VCF to allow other filter types and sounds to be added to the repertoire of electronic musical instruments. We will explore in detail some of the responses the 3350 is capable of generating, present the details of how they may be implemented, and attempt



The Current Curtis Line.

These devices have become the workhorses of the industry. All still actively used by major manufacturers, NOT one chip has obsoleted an earlier design.

Interview with Dave Smith of Sequential Circuits: Creating a Market with Logical Instrument Design.

Sequential Circuits does not fit the mold of the many high tech companies in world famous Silicon Valley. Classical music plays in the hallways. Wall-sized super graphics proclaim the company's involvement in music and creativity. While the atmosphere is casual and most people seem to be well under age 35, there is also a feeling of extra-ordinary dedication, energy and creative teamwork extending well beyond the 9-5 work span.

S: Since the 1979 CK article that I read Sequential has introduced programmable FX, the Pro 1 and Ten voice. And what's next?

D: That's all we're talking about now. The FX will be the next big thing. We didn't really announce that to the public, it was just introduced to the industry at the (Chicago

Continued on page 8

to give some idea of how they sound.

The CEM 3350 Dual VCF

For those of you who are not entirely familiar with the 3350, here is a brief review: The 3350 contains two completely independent state variable filters as shown in figure 1. Each state variable has a fixed gain input, V_{IF}, where the gain is constant in the passband as the Q is varied, and a variable gain input, V_{IV}, where the pass-band gain decreases as the Q is increased. Each filter has two outputs, a low pass response, V_{LP}, and band-pass response, V_{BP}, appearing across the two filter capacitors; these outputs normally must be buffered and amplified by an external op amp. A high pass response is also possible by coupling the input into the V_{BP} pin, and taking the output from across the V_{BP} capacitor (appearing at the V_{LP} pin is now a band pass response). Finally, each filter has independent frequency control and Q control, both with standard exponential scale.

The real power and versatility of this device is that the two state variable filters may be interconnected in a wide variety of ways to produce many different responses. But even more importantly, there are a total of four independent parameters which may be controlled in some relationship to each other, thus allowing

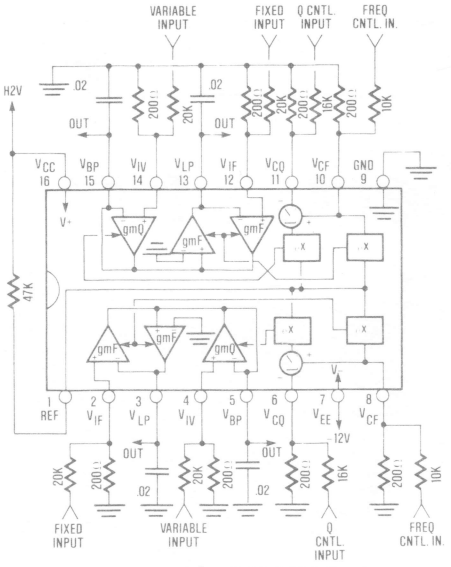


Figure 1: The CEM 3350 Dual V.C. State Variable Filter

the responses to be dynamically varied with even greater variety.

An Evaluation Circuit

Figure 2 shows a simplified block diagram of a test circuit which may be used to explore the various interconnections of the two filters. Note that for each half, one switch applies the input signal to either the normal fixed gain input or the the V_{BP} pin for a high pass response, while another switch selects between the two outputs. Thus a 2-pole low pass response is obtained with both these switches in the up position, a 2-pole high pass is the result of both switches in the down position, and a band pass response is outputed when the switches are in either of the two remaining combinations. Finally, the series-parallel switch connects the selected response of VCF #2 either in series with or parallel with the selected response of VCF #1.

This test circuit also provides the two most obvious ways of controlling the frequencies of both filters in some relationship to each other. With the separation control input, F_S set to zero volts, the two Fc control inputs will receive the exact same voltage; thus, the two filter frequencies will be identical and will track each other as the tracking control input, F_T, is swept. A positive voltage applied to the separation input, on the other hand, will sweep the frequency

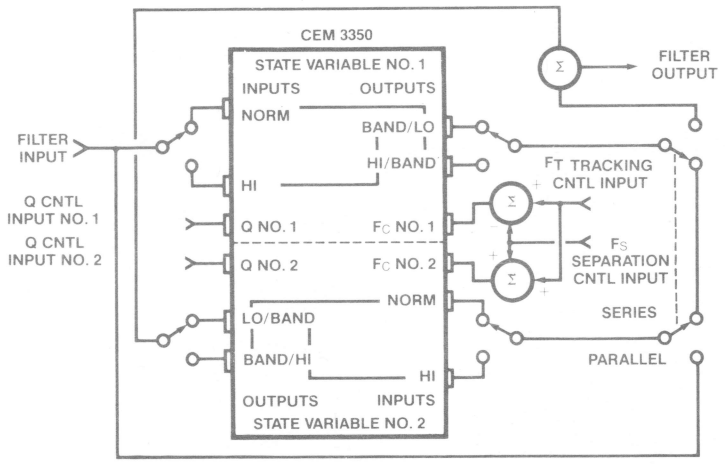


Figure 2: Block Diagram of CEM 3350 with Switch Selectable Responses

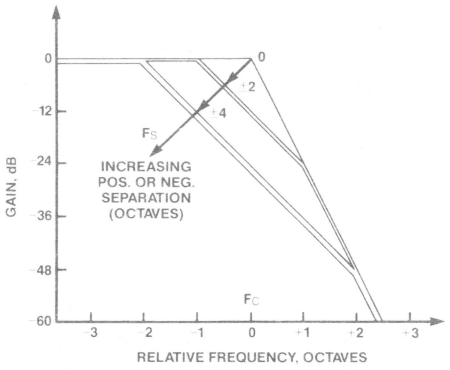
of VCF #2 higher while pushing that of VCF #1 lower. Note that a negative voltage for F_S will also separate the two frequencies, but in the opposite direction. If the tracking input is now swept with a non zero F_S the two frequencies will again track each other, but with some constant interval between them.

Some Typical Responses

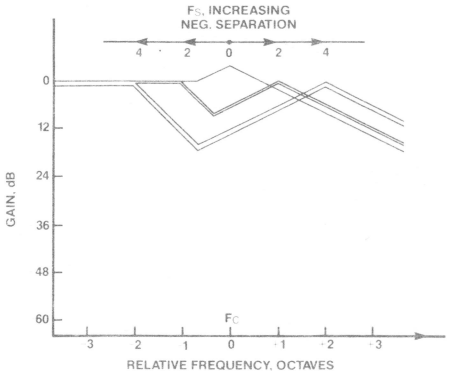
Figure 3 shows Bode plots of several of the possible responses. Each different line type represents the response at various settings of positive and negative separation with the tracking control held constant. The amount of separation is shown as the total interval in octaves between the two filter frequencies. A curve marked +2, for instance, would result from a positive voltage applied to F_S causing filter #2 to be raised 1 octave and filter #1 lowered 1 octave. Also shown on these plots is the movement, or progression, of the shapes as F_S, for example, starts at a negative value, goes through zero, and increases positive.

For the sake of clarity, the responses are shown with the Qs of both filters set at less than unity. Increasing one or both of the Qs will put resonant peaks at the respective one or both of the inflection points in the responses.

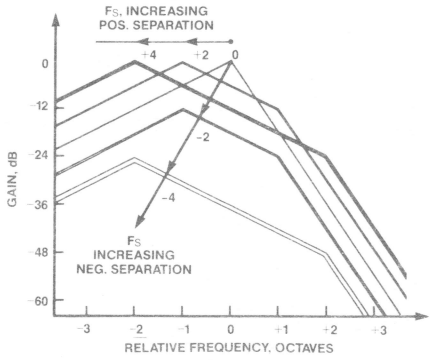
Upon study of these plots, a number of most interesting points become apparent. First of all, the general effect of the separation control input is to change the basic shape of the response, while the tracking control input simply moves the whole response up and down. This is in sharp contrast to our standard filter, which always has the same response shape-- namely



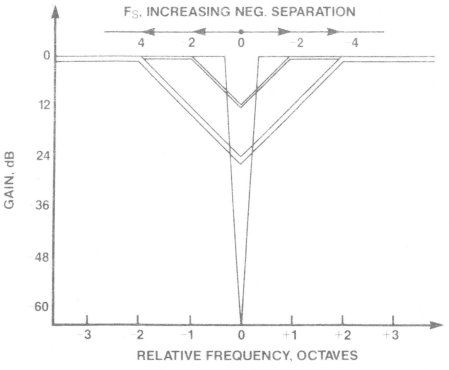
Response of Low-Pass/Low-Pass, Series Connection



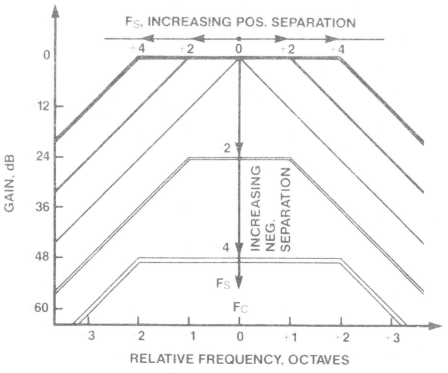
Response of Band-Pass/Low-Pass, Parallel Connection



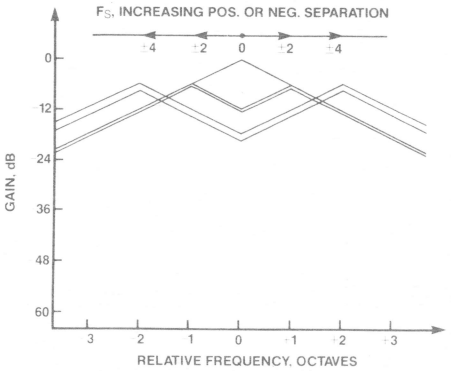
Response of Band-Pass/Low-Pass, Series Connection



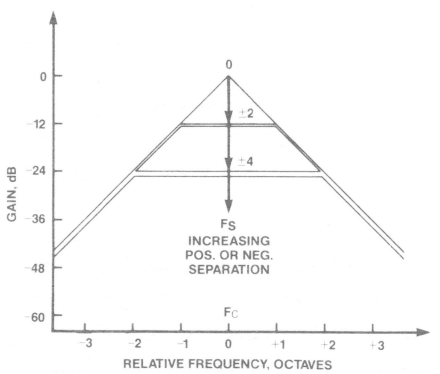
Response of High-Pass/Low-Pass, Parallel Connection



Response of High-Pass/Low-Pass, Series Connection



Response of Band-Pass/Band-Pass, Parallel Connection



Response of Band-Pass/Band-Pass, Series Connection

Figure 3: Sample CEM 3350 Responses

that of a low pass.

Another important observation is that for some series connections, increasing the separation in one direction not only changes the response shape, but lowers the overall amplitude of the signal. For instance, with the high-pass/low-pass series connection, an increasing positive separation simply increases the bandwidth; an increasing negative separation, however, in addition to increasing the bandwidth, also lowers the gain of the filter at a rate of 12 dB for every one octave of separation.

This is an important result, because filters can also double as VCAs. A common patch is to envelope sweep a low pass with the initial frequency of the filter set below the fundamental of the tone. Normally, then, no sound comes through; but when keyed, the filter opens up, enveloping the tones loudness as well as its filtration. The only problem is that this amplitude modulation is limited to occuring only on the low end of the sweep, at the point when all the harmonics have been filtered out and only the fundamental remains. With the above band pass response, however, the amplitude modulation can be made to occur at any point in the tone's spectrum; and the spectrum actually increases as its loudness is decreased. The point is that this filter offers a completely different flow or progression of the dynamics when modulated than our standard 4PLP-- and hopefully the resulting sound is just as musically useful.

A few more general comments should be made: When the response selected for each state variable filter is the same, the effect of separation will be symetrical about zero F_S; that is, positive and negative separation will produce the same result. Also note that in the series connection, the order of the selected responses makes no difference, other than to reverse the effects of positive going and negative going separation. A band-pass followed by a low-pass will produce the same responses as a low-pass followed by a band pass. Of course, it also makes no difference when connected in parallel.

Finally, we should point out that the resulting response of two responses in series will have somewhere cut-off slopes equal to the sum of the individual slopes, whereas with the parallel connection, the resulting slope will be approximately equal to the lessor of the two individual slopes. In other words, a 12 dB per octave slope and 6 dB per octave slope in series will produce a 18 dB/octave slope, but only roughly a 6 dB/octave slope when placed in parallel.

Details of Implementation

An actual schematic of the test configuration discussed above and shown in the block diagram is shown in Figure 4. To allow ease of investigating all responses and interconnections, each output of the two filters has been buffered separately with a BIFET op amp with gain of 100. The input signal is either connected to the fixed gain input for low-pass and band-pass responses or capacitively coupled into the V_{BP} pin for high-pass and band-pass responses. Also included for each filter is the diode feedback network for preventing "jump resonance" at high Q settings and high input levels (see CEM 3350 data sheet). Provision has been made via jumpers to apply a small portion of each bandpass output to its fixed gain input; this feedback connection will cause that filter half to break into a pure sine-wave oscillation at a high Q setting.

The control section includes op amps A5 and A6, one performing frequency tracking and the other frequency separation. In addition, there is a separate control input to each frequency control pin to provide yet another way of controlling the frequencies of the two filters in some relationship to each other: namely, one frequency is kept stationary while the other is swept above, below, or through it. Finally, each Q control pin has its own control input.

For a specific filter application which does not require all the response and interconnection

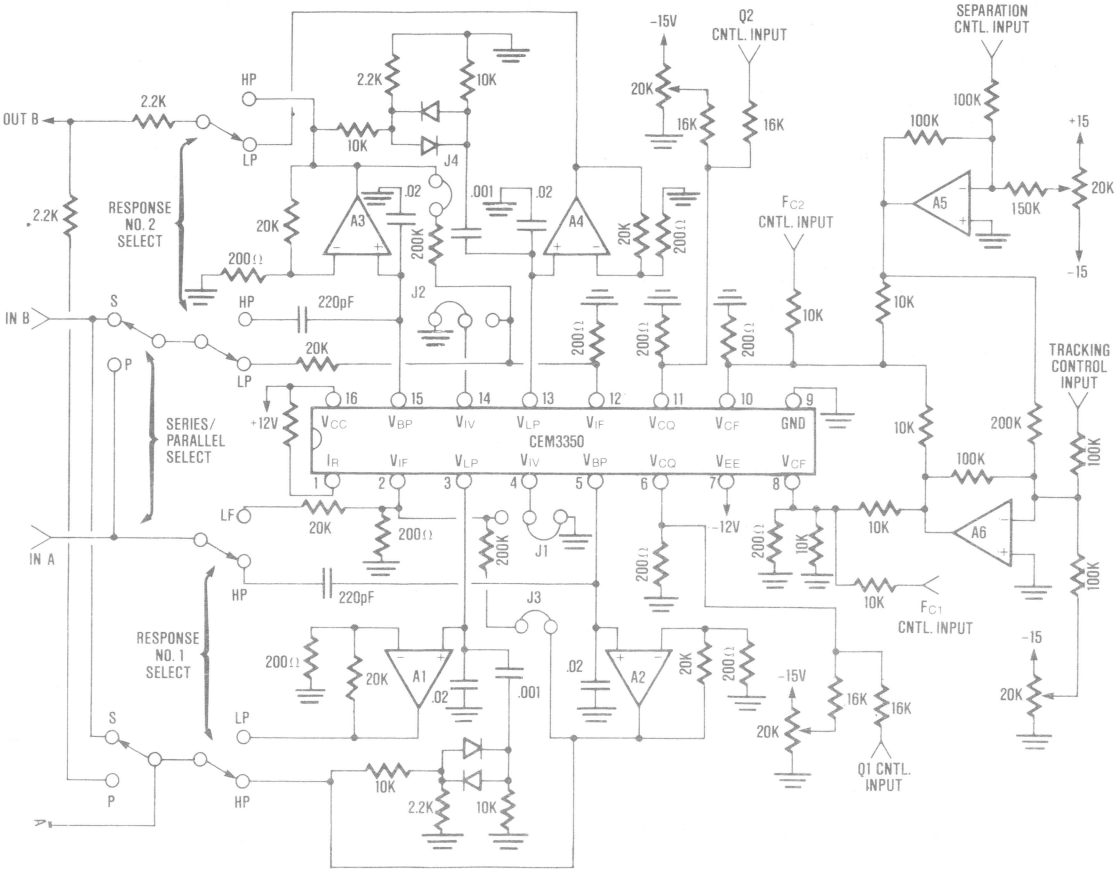


Figure 4:CEM 3350 Evaluation Circuit

possibilities of this circuit, substantial simplifications and reductions in external circuitry can usually be made. For instance, suppose only the high-pass/low-pass series connection is desired with tracking control, separation control, and a single Q control; then only the external circuitry shown in Figure 5 would be required A practical circuit of a standard 4 pole low pass which can be adjusted to a bandpass with 6 dB/octave rising slope and 18 dB per octave falling slope is shown in Figure 6. The CEM 3350 data sheet should be consulted for further details.

Hearing is Believing

So, what do all these new filter configurations sound like? We normally hate to make any subjective interpretations for fear of limiting or influencing what someone else may hear in the same sounds. But in this case, we feel some description may be useful.

In general, the series interconnections seem to be more dramatic when swept than the parallel interconnection. This may be due in part to the larger amplitude changes that usually occur with the spectrum changes in the series connection. Or it may be due to the generally steeper slopes and correspondingly greater phase changes which occur during modulation. The parallel connection, on the other hand, seems to let more of the input through while still filtering it, giving the sound more presence, and the sweeps are smoother.

Also, the difference between the various combinations seems to relate directly to the basic sonic differences between low-pass, bandpass, and high-pass. The low-pass has a full, bassy sound due to the predominance of the fundamental and lower harmonics. The bandpass sounds like the low-pass, but with the bottom end dropped out, giving a more hollow, midrange sound. The

high-pass, because it lets only the upper harmonics through, has a very bright and brilliant sound.

The combinations of these basic responses seem to take on the same characteristics as they have individually, while providing intermediate blends between them. For instance, a low-pass/band-pass combination is not quite as full as a low-pass/low-pass, but has more bottom end than a bandpass/bandpass. And a high-pass/bandpass has more brillance than a high-pass/low-pass, which itself has a more hollow sound than a bandpass/low-pass. And so on. The result is a great deal more subtlety of flavors to choose from.

Probably the most important aspect of these filter combinations, however, is how they are swept. Let us start with the low-pass/low-pass series combination. With the separation at zero, we have essentially a 4 pole low pass, which when swept, sounds pretty much like any other 4PLP. One difference is we can get quite a bit more resonance, if we wish, because higher Qs are possible with state variables and because both resonances are aiding each other.

If we now separate the two filters slightly with the separation control, and sweep the tracking input, we get quite a noticeable difference, especially with both Qs cranked up somewhat. In our opinion, the sound is much fatter and more interesting than with the two frequencies perfectly coincident, similiar to the increased richness obtained when going from one oscillator to two or three oscillators in parallel.

An interesting thing also happens when we rapidly sweep the tracking input up and down with slight separation: The sound takes on a very reverberant quality. And as we increase the separation, the reverberation seems to become longer and longer, finally turning into more of an echo. We suspect this effect is due to the resonant peaks and their corresponding steep phase transitions following one after another during the tracking sweep.

So far we have just been sweeping the tracking control input. What happens if we now sweep only the separation control input? The results are even more astonishing, sounding nothing like what we are used to hearing when a filter is swept. The resulting sound has a rich, human vocal quality, not unlike the sound of vowels being spoken. Upon thinking

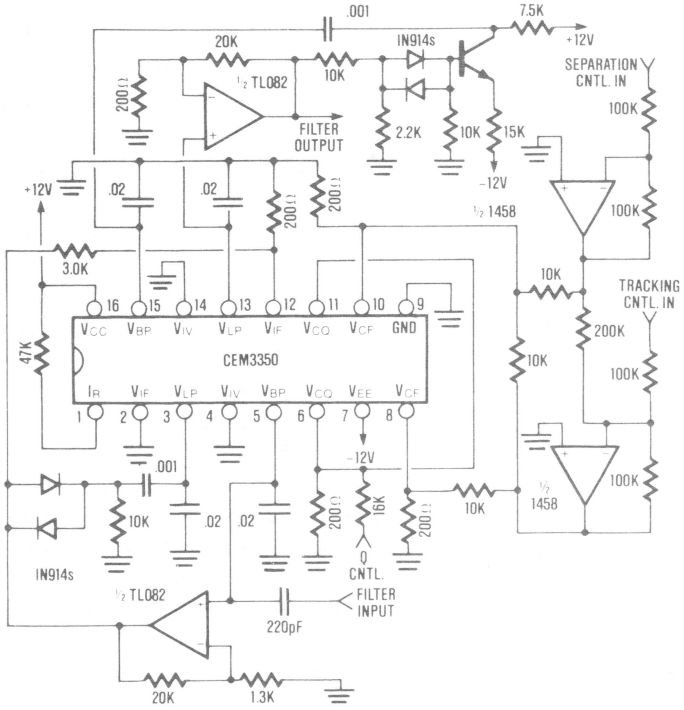


Figure 5: Practical Circuit of Variable Bandpass

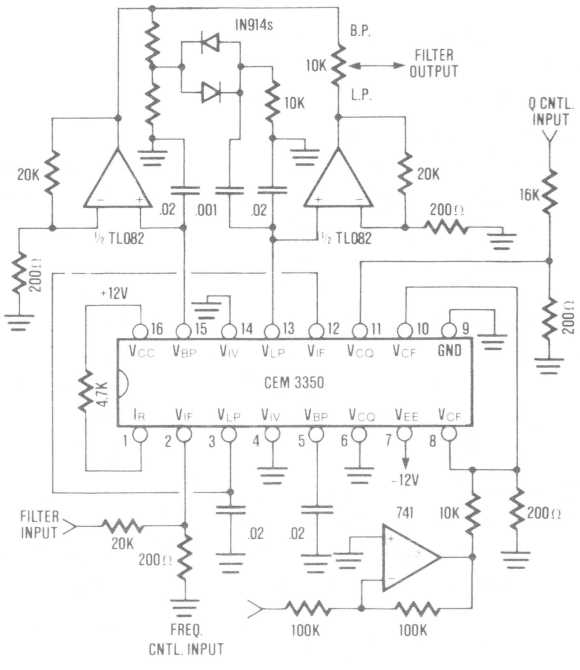


Figure 6: Practical Circuit of 4 Pole Low-Pass/Bandpass

about it, the effect is not really too surprising, since the human vocal tract is made up of multiple resonant cavities which we move in some relationship to each other - in this case apparently in opposite directions - in order to form our vocal sounds. It is exciting to us to think of the musical applications of such vocal sounds.

It is also interesting to note the tremendous difference in sound from just sweeping the same filter configuration in two different ways -- one tracking, and the other separation. This leads us to think that the particular relationship used in controlling the frequencies of the two filters may be even more important in forming the characteristics, especially the dynamic progression, of the sound than the actual response shape or combination of responses. In fact, the qualities we just described for the tracking sweep do also occur to a greater or lessor extent in all filter configurations, regardless of response combination and interconnection; these qualities are only modified by the blend of the basic characteristics of low-pass, bandpass, and high-pass as discussed earlier.

If this theory is correct, then the possibilities for unique filter sounds are limitless, and the concept of controlling two or more independent filters in some relationship to each other becomes a very powerful subtractive synthesis tool. Please note we said two or more, because there is nothing stopping us from adding another 3350 in series or parallel, or combination thereof, and having four filters to control in some manner. As we said, the possibilities are limitless.

Some Other Applications

Before we conclude this article, we should point out some applications for the 3350 other than dual VCFs. As mentioned earlier, each state variable can be made to oscillate with a feedback resistor between the proper points. The two outputs will be fairly pure quadrature sine waves whose frequency can be exponentially voltage controlled via the frequency control pin, V_{CF}. The scale accuracy is excellent for the lower 6 or 7 octaves, and the sine wave amplitude remains very constant over the entire range. Thus either filter section can also double as a wide range sine wave VCO. One interesting application is to operate one section as a VCO and the other section as a VCF. One VCO output can be converted to a variable pulse with a comparator; then by adding simple envelope control, we have a simple one chip voice suitable for less critical applications.

Often an application may require a simple 2 pole filter and a VCA. The versatile 3350 can also fill this need: Figure 7 shows how a section may be operated as a VCA by putting -0.2 to -0.5 volts on the Q control pin to turn-off the Q transconductor, grounding the V_{LP} pin, applying the input to the fixed gain input, and taking the output from the V_{BP} pin (which must be fed into a virtual ground summing mode). The frequency control pin, V_{CF}, then controls the gain; this pin should be operated between -50mV and +200mV for an approximate linear scale and between +50mV and +300mV for an exponential scale. For lower feedthrough when operating in the linear region of the scale, the reference current, I_{REF}, should be set between 100 and 200uA.

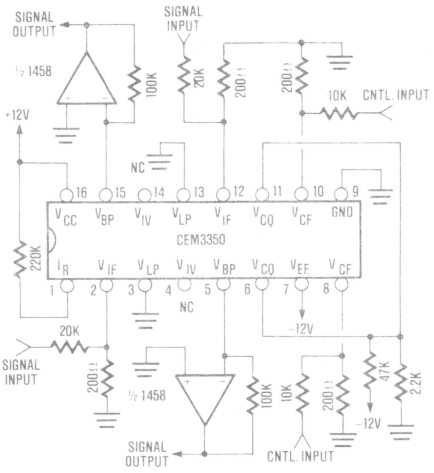


Figure 7: Gain Control Using the CEM 3350

Conclusion

As you can see, the CEM 3350 is a pretty flexible device. Its greatest asset is its ability to generate new and rich sounds for musical instruments. We hope that after you have experimented with its many possibilities, you will share your discoveries with us as we have shared ours with you.

NAMM) show. We have also recently introduced a couple of accessories -- the polyphonic sequencer for the Five voice and the remote keyboard, also for the Five voice. They are actually going to be the next two products on the market. Those will both be in production by the end of the year. The FX will just be starting up at the beginning of next year for actual production; and because it's modular, we can keep introducing new modules periodically. We are working on a few other things, but nothing that we would like to comment on right now.

S: How was the FX received at the NAMM show?

D: It was a mixed reception. People who were able to comprehend and understand what it was liked it immediately, and of course, had a lot of questions about what the instrument did. We found that a number of people didn't understand what it was. If we had a guitar plugged in, they thought it was a guitar synthesizer; if we had a violin plugged in, they thought it was a violin synthesizer, since the range of things we did, even with the few effects we had, gave the ability to change the sounds dramatically and easily. Whenever you do something like that, involving a lot of coloration of the sound, people think it must be a synthesizer, as opposed to just standard processing modules, as the effects are.

S: Prophet 5 is what we would call a fairly mature product, introduced four years ago, and you've made a lot of re-designs, revisions, and improvements. Can you describe these, and tell us what brought it about.

D: I suppose right now that there are four major versions of the Five voice. There's the original one which we refer to as Rev. 1. We built about 180 of those. There's a second version which we call Rev. 2, which was only a partial re-design, but it did incorporate a lot of improvements that made it quite a bit better. There are probably about a thousand of those out there. Both of those versions incorporated the (other source's) chips. Rev. 3, which we have manufactured the most of, was introduced about a year and a half ago. That unit incorporates a complete redesign of the internal circuitry of the instrument. We switched over to using the Curtis parts as well as a number of other innovations in the circuitry to make it more reliable, more produceable, cleaner, more in tune, etc. Just a better designed instrument than before. There's one more version since then that has been out for about six months now; the only real difference in outward appearance is that there are two extra jacks in the back that provide the interface needed for the sequencer and remote keyboard. It is considered a small refinement, rather than a major change.

Sequential Circuits President, Dave Smith (left)

S: What's the story behind the switch to the Curtis chips?

D: We were real small when the Five first came out. We only had about five people, and because of the Prophet's popularity, we were growing quickly. Trying to overcome parts problems, trying to produce these things, trying to grow, and do all these things at one time didn't leave us time to breathe. We really didn't have time for someone to sit down and redesign it as early as we would have liked. We tried a few times, but it would keep getting put off and

put off. Meanwhile, Doug would stop by every once in a while, tell us about his latest parts. It finally got to the point where he got his oscillator done, and he already had the filter and the transient generator. At about the same time, RCA came out with a part that we decided to use as a VCA. All of that sort of happened at one time. And we were also, even 2½ years later, still having problems with (other chip source), so we just decided that our highest priority was to do the re-design, get rid of the old parts, design in the new ones and start working with them. Doug lent us a hand with that, and spent some time helping us get them going. I would say ever since, we've not had one problem in delivery or reliability because of Synthesizer chip sets, whereas before it was a daily occurence. He came out with the parts just in time for us, and it didn't take a whole lot of talking on his part for us to switch.

S: What reason do you get from your marketplace, what's your selling edge, why do musicians buy Sequential products?

D: Well, there are a lot of reasons. Mainly reputation at this point. Timing is also a factor. Since we were the first ones to come out with this type of instrument, we had about a year and a half by ourselves. We picked up a real good following immediately from the professionals. When you get a big acceptance from professional people, it filters down. Looking at warranty card demographics, users are asked why they bought it, and usually it's quality, reputation, state-of-the-art, etc. People make statements like -- "Everytime I saw someone come into town, or see anybody on T.V., they have a Prophet" -- things like that...I think it's a combination of all these things. Plus, we started out building state-of-the-art higher cost products, and that tends to give a company a higher image than if you built a lot of low cost products. For instance, if we had built the Pro-1 first, then the Five voice, that probably would have been different than if we had built the Five voice first, then the Pro-1 afterwards.

S: What musicians influenced the product design of your product? The trend of the product design? Specific musicians or musical trends.

D: To be honest, we really haven't gotten too much specific input from musicians that come to us like a blinding light or revelation...You can ask them, "What would you like to see in a synthesizer?" and usually they'll stop cold in their tracks, because they don't really know. When you come out with something, they can say you should have had this, and you don't need that. But, not that many people really understand synthesizers enough to really know what they want. Yet, some want some things that are so esoteric or specialized as to not be useful to the

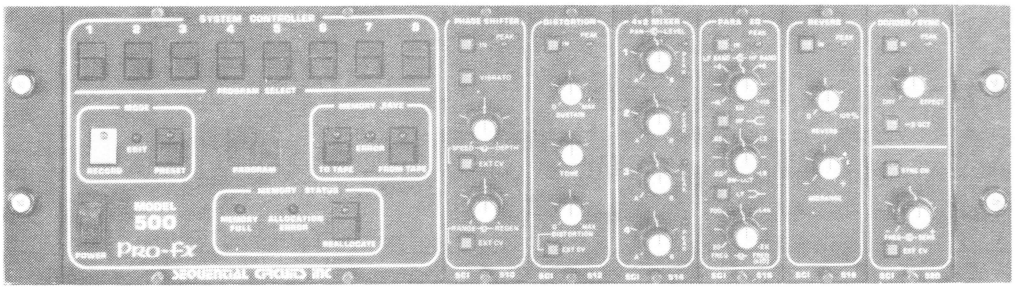
TO OUR READERS:

We apologize for being so late in delivering our second issue. This was supposed to be our Spring issue, but instead of playing "catch-up" for the next several years, we decided to combine Spring and Summer into a single issue which does offer the compensation of being considerably longer.

The response to our first issue has been overwhelmingly favorable, and we are pleased that many are benefiting from our newsletter. Thank you for your feedback, support, and patience.

About the only criticism we have received is that some of the articles contain sales pitch overtones and that most articles have to do with our devices. In our defense, we must admit that one of the main functions of Synthesource is to disseminate company related news and provide additional applications of our products. So, the articles describing our new products are bound to sound a little like a salesman talking. And when we discover new applications for our devices, we can't help it if we get enthusiastic -- the chips are flexible (note all the applications for the VCO chip presented in this issue) and they are good. However, we promise to make an effort in the future to tone ourselves down. Also, we remind you that we do encourage circuit ideas and articles on topics other than our chips.

So with that, we now turn you over to our second issue.



The Programmable FX From Sequential Circuits

mainstream musicians.

S: How do you think the people mix at Sequential has affected your product line?

D: I think it helps. The fact that we do most of our design in-house, testifies to the capability of everyone here to come up with original suggestions. Like the Pro-1, I personally didn't have quite as much to do with that as the Five. Of course, I was the only technical person here during the development of the Five, so I had to do all that myself. But, some of our other designers, and the people in sales all had input on the Pro-1 and it was a combination of everybody putting ideas together and deciding actually what was going to be on it. We have some standard disagreement between sales and engineering, because engineering thinks one thing and sales will think another. Sales of course, wants to see something that does everything and costs nothing, and the engineering staff has to get the reasonableness in there of cost versus performance, which ends up being one of the most important parts in the design. I think a lot of people overlook that.

It's hard to decide where to draw the line in development sometimes -- especially on products like sequencers and synthesizers. They can literally do anything if you want to design something in -- there's always another feature you can add. But you have to be able to say this is where we draw the line, because 3% might use that -- it's just not worth doing. But this one is useful, because 50% of the people use that. And that's where the arguments come in...that's when a catalyst of people coming from different directions really helps.

Bob Styles, our Sales Manager, is from England, and he has a lot of experience. He was a professional keyboard player and managed a keyboard store over there. He not only has a lot of input from running a keyboard store, but also as a musician working with a lot of the top bands in England.

Kevin Kent (also in Sales), had a lot of schooling in music, plays everything from flute and saxophone to piano and synthesizers, and has done a lot of studio work. He also worked at a Guitar Center in L.A., which gave him first hand experience seeing how salesman push what, what sells and why. Greg, who is also in the Sales Marketing Department, has a degree in music, and is a very good piano player.

People in our Engineering Department -- Chet Wood, our Engineering Manager, has a Master's in Music Composition from Mills College. He has been working with synthesizers for quite awhile. Steve Salani, who ended up doing most of the final design on the Pro-1, has been dickering with synthesizers for at least 10 years, and is an excellent pianist as well as synthesizer player. He has a large modular synthesizer he designed for himself. Our entire engineering staff has synthesizer, keyboard, musical background, as well as engineering and design capability. And everybody is coming from a slightly different direction, with a slightly different perspective.

S: You identified yourself as a quality product, yet the bottom line is how it sounds.

D: That's true. And that's one of those funny things, because there are two things that affect the sound. One is just what set of front panel functions you choose to use, which pretty much defines how

much control you have over the sound and how much variation is possible. This you do have control over. The second determination is the electronics you use; which, in a way, you don't really have any control over. Bob Moog said the same thing about the Mini-Moog. Ask him why it sounds good, and he'll probably say, "Is it the filter?; maybe, probably, I don't know." On paper, two circuits can be identical. For example, filters are usually 24 dB low-pass filters, but they can sound different. When we built the first Prophets and even when we redesigned it, much as I would like to say otherwise, I had very little control over the sound quality of the electronics used. I had a control over what knobs and switches went on the synthesizer to give the wide range of sounds you can get on a Five.

We did have to choose which actual circuits to use -- and this can partially determine the "sound" of the instrument. In fact, it was a little scary when we swithced to Curtis chips because it was a new oscillator, new filter, and new transient generator. We did a lot of looking and listening, compared our standard factory presets (before and after), and so forth; and they do sound different. I mean, I can't deny the fact that earlier Prophet's sound different than the newer ones. Some people say the old ones sound better, some people say the new ones sound better. I think that some people tend to think that anything that's older has to be better, so they say it's the older ones that are better just because they are older. Lots of people see that in other musical instruments. You hear all the pre-CBS business of "it's older so it has to be better." I personally would prefer the newer ones because they work better, they are more reliable.

S: What elements do you think are most responsible for differences in how different instruments sound?

D: On the newer units, the filters and envelope generators are a little different from the earlier units. The old instruments, because they're more out of tune and sloppier, had more noise, and more distortion, and had more coloration. People sometimes confuse a full sound with oscillators being out of tune, whereas other people screech if the oscillators are the slightest bit out of tune. The new ones are hardly out of tune at all. The old ones are usually out of tune, but if you hit a big chord, all the oscillators will beat unevenly because they are out of tune, and it sounded full. So, it's a combination of all these things. We've had people bring a machine in, with a lot of distortions and things, because the instrument was out of tune, and our Service Department would fix it all, but then the guy wouldn't like it. He liked all the little distortions and things that were going on. So, you reach a real subjective point.

S: What would your dream synthesizer be?

D: My dream synthesizer. Probably some sort of modular programmable polyphonic unit. I don't even know how it would be packaged. It would allow you to do everything you can do with a modular system as far as unlimited patching, whereas something like a Prophet-5, or some of the other synthesizers that are out now, allow you to select only from signal and control paths that are available in the instrument. So, with this type of instrument you can put in as many kinds of modules as you want and program the interconnections, and if it was also polyphonic on top of that, then you could do anything that a modular could do, plus have the advantages of a programmable polyphonic unit. You'd get the best of both worlds. Unfortunately, it would cost too much and would be too hard to understand to make it a viable commercial product at this time.

S: Would you go so far as to say that music hasn't kept up with the technology?

D: I think that has always been true to a point. There's always been so few people who attempted creativity -- even with the Prophet. There's so many things you can do with it--people working at Sequential are always finding new sounds. And yet, most people don't even try.

S: Going to the other extreme, what do you think the weaknesses are in the

synthesizers out on the market today?

D: It's really hard to say what the weaknesses are in the synthesizer themselves. Because, again, I feel that something like a Prophet-5 has so much capability that people aren't using, it's hard to say there is a significant weakness. One problem a synthesizer can have is whether it's too hard to understand or not. And that's one thing that people have said about Prophets, that they are very easy to understand if you've ever worked with any kind of synthesizer. We purposely kept the layout very simple and organized. You can take this problem to the extreme when you start talking about digital synthesizers. They have a major problem with the user interface in general. Affording the high cost isn't valid to argue about, because eventually they're going to become less expensive. Technology is going to take care of that. I tend to not worry about what they will cost, but rather how they are set up. I think people are finding that the only way they're going to be viable is in completely preset instruments. One company has just introduced a $5,000.00 digital machine that offers no user programmability, which is back to the old game of being stuck with someone else's presets. Even though people don't change their patches often, I think they will hesitate to purchase a synthesizer if they know they can't ever change their patches.

Setting up sounds on your own on a digital synthesizer is extremely difficult and time consuming. There is nothing technology can do to change this.

S: So, what's the future of the digital?

D: I think it's going to be in preset machines...I personally don't think there's ever going to be a mainstream interest in programmable digitals unless they change their whole architecture. If somebody emulated an analog instrument in digital, then that would make sense -- where you have digital oscillators, digital filters, digital envelopes generators, and your front panel looks just like a standard synthesizer and acts the same way, only it happens to be digital. There is probably a market for that sort of thing when it gets inexpensive enough. But beyond that, the way they are set up now with additive syntheses and FM and so forth, it's nearly impossible to set up sounds. And as such, is near useless on a practical level.

S: In 1979, you basically forecasted the CASIO product line. Did you know about it at the time, and what do you think of it now?

D: I hadn't heard about them at that time. Now, CASIO is an example of using digital technology in a good way, because those are totally digital and it's all preset, specific, and inexpensive. It does a lot for the money. There's nothing wrong with those kind of instruments. They don't affect us because they are not in the professional market at all. In addition, they are not really synthesizers in the classical sense. They're organs.

S: How has that affected the state of the art? Has it just fallen into another category?

D: I think so. I think they've done a good job in creating a whole new market. It's more or less like we did with the Five voice. Before the Five voice came out, people used to say that the synthesizer market is declining; it's peaked, and that was it. Of course, we haven't done too bad with the Prophet-5 by essentially creating a whole new market. After that, we heard that monophonic synthesizers were dead, they've been out for 10 years and no one wants them anymore. With the Pro-1, I think we've also proved that to be quite the opposite of what people imagined. It's a matter of getting a logical instrument at a logical price and creating your market. I think that's exactly what CASIO's done.

S: What's your feeling about competition in general?

D: I tend to not worry about it too much. I'm not the kind of person who sits around and sees who does what and worry about it. I'd rather spend my energy on developing innovative new products. I sometimes see in the industry people running scared of this and that. Build something logical, something that's good, and you won't have to worry about the competition. There's no magic involved.

READER'S CIRCUIT CENTER

Welcome to our very first column featuring circuit ideas from our readers. We are very happy to be able to open this column after only the first issue. Thank you for your response.

The honor of the first circuit idea for this column goes to John McColm of La Grande, Oregon. His circuit for gating the CEM 3340 VCO on and off reminded us of the trick with the hard sync input that we had long forgotten about. This input, which is pin 6 on the 3340, is connected to an internal bias point on the switching comparator which is fed from a -2.5 volt source through a 6.8K resistor (see the CEM 3340 data sheet or previous issue of Synthesource). The original purpose of this pin is to allow the state of the comparator to be set with pulses capacitively coupled into this pin, thus producing hard synchronization effects (see also "Application Corner" of this issue). However, by forcing this pin to a voltage at least one volt above or below the -2.5 volt bias, the reference input to the comparator can be forced continuously to either ground or the internal 2:1 resistor divider. With the positive feedback around the comparator now effectively disabled, the negative feedback around the oscillator loop will cause the triangle output to stabilize at a level which balances the internal comparator -- namely at either 0 volts or at the resistor divider voltage, depending on how the comparator has been forced. The resistor divider voltage, called V_{REF}, is nominally 1/3 of the supply and appears at pin 9 on the 3340. Table 1 gives the steady state values of the triangle, sawtooth, and pulse outputs versus the level to which pin 6 has been forced. As can be seen, forcing the hard sync input pin to either the higher or lower level will not only stop the oscillator from oscillating but will force the triangle and sawtooth outputs to a well determined and known voltage.

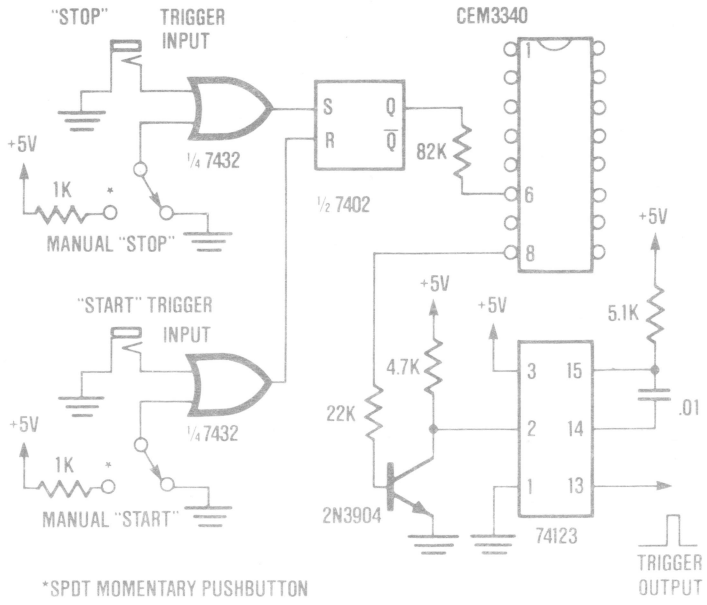


Figure 1: Gateable V.C. Clock

HARD SYNC INPUT	TRIANGLE OUTPUT	SAWTOOTH OUTPUT	PULSE OUTPUT [2]
-1.5 to -0.7 V	0V	2·V_{REF} [1]	LO
-3.4 to -4.2 V	V_{REF} [1]	V_{REF} [1]	HI

NOTES: 1. V_{REF} = +5V @ V_{CC} = +15 V
2. P.W.M. INPUT = 1/4 V_{CC}

Table 1: Output Levels During Gating

John McColm made use of this trick for gating the 3340 in his circuit, shown in Figure 1. John writes, "This is a circuit that gates the VCO in response to external TTL level triggers or pushbuttons. I use it in my system as a voltage controlled clock to start and stop a sequencer. For clarity I've ommitted the normal outboard circuitry for the 3340, and shown only the trigger input and output structures that I use in my application. Perhaps others will find the circuit as useful and versatile as I have." We should mention that John did not have the benefit of the foregoing discussion and came upon this idea himself.

Although forcing pin 6 high with a single resistor from the logic output is acceptable, it does alter the internal bias level somewhat even with the

gating logic at the low level. This may cause erratic or no oscillation for some value of supplies, and so the value of this resistor will have to be adjusted accordingly. Figure 2 shows methods of interfacing logic to the hard sync pin which are less critical because they affect the internal bias level only when gated; diagram "a" may be used for forcing the pin to the high level with a high logic signal, while diagram "b" shows how to force the pin to a low level with a low logic signal. For best operation, the forced levels should be kept within the values shown in Table I; Figure 2 gives the necessary design equations for maintaining these conditions.

The fact that this gating technique forces the triangle and sawtooth outputs to well defined levels when the oscillator is gated off may be used to advantage. For instance, the triangle waveform can be made to start from its lowest level or from its highest level; the sawtooth can be made to begin at its midpoint or at its highest level. These features make the VCO useful for gated modulation purposes. Or the waveforms may be made to begin at any level by forcing the V_{REF} pin (pin 9 on the 3340) to any desired value between 0 and +5 volts during the gate off period. (The resistor divider on the V_{REF} pin may be over-ridden with the low impedance output of an op amp.) A typical applications would be in generating a tone burst signal, where the waveforms should be started at their midpoint.

As long as we are discussing gating techniques, we should point out one additional way of gating the CEM 3340 -- that produced by forcing the V_{REF} pin to ground. The advantage of this method is that it forces both the triangle output and the sawtooth output to the beginning of their cycles or at zero volts, making this technique ideally suited for modulation purposes. A practical circuit using this method is shown in Figure 3, which was submitted, we admit, by ourselves.

In this application, the triangle and sawtooth outputs are normally sitting stationary at zero volts; upon a momentary closure of S1 to V_{DD}, or a positive transistion occuring at the external trigger input, these outputs will cycle through one to nine complete waveforms, depending upon the setting of the thumbwheel switch, and then return to their initial condition.

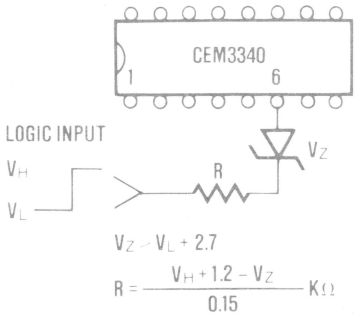


Figure 2a: Gating on High Level

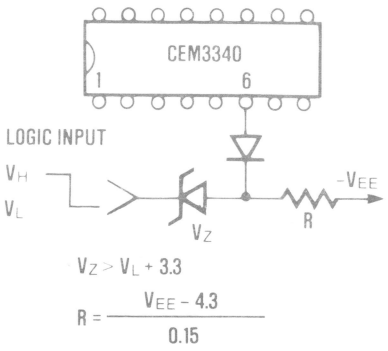


Figure 2b: Gating on Low Level

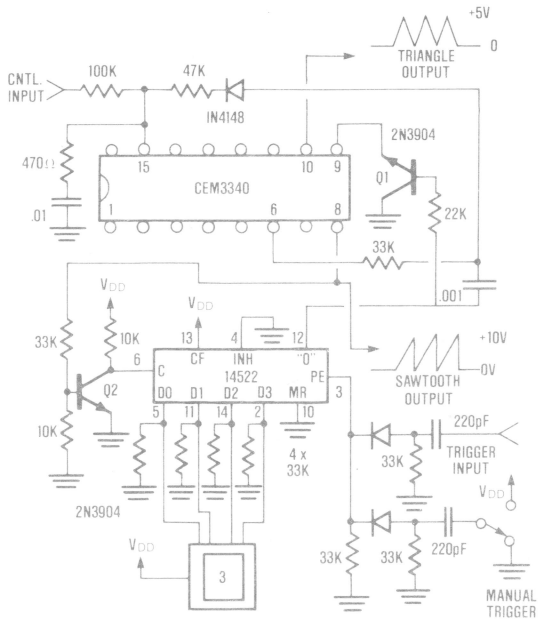


Figure 3: Programmable Variable Cycle LFO

This programmable feature is accomplished by the 4522 CMOS programmable down counter. The start pulse loads the desired count into the 4522, causing the "O" output to go low, which turns Q1 off and allows V_{REF} to return to its normal value. The 3340 begins to oscillate, decrementing the counter on every negative transistion of the sawtooth wave. When the 0 count is reached, Q1 turns back on, gating the oscillator off in the start position of the waveforms.

Transistor Q1 is operated in its inverted mode, so that when turned on, it pulls V_{REF} down to within 10-20 mV of ground. The remaining network associated with Q1 has several purposes: Whenever the oscillator is gated off, a positive pulse is coupled into the frequency control input; this prevents an internal parasitic effect from latching up the oscillator. Whenever the oscillator is gated on, a negative pulse is coupled into the hard sync input; this sets the comparator in the proper state necessary for preventing a momentary positive pulse from appearing at the sawtooth output. (Positive pulses coupled into this pin are not necessary, but have no adverse effect.)

This, then, concludes our first reader's column. Please keep those ideas coming, because as you can see, they usually turn out to be very useful and constructive (see the box for some additional incentive).

THEORY & PRACTICE

Inside Workings of Voltage Controlled ADSR'S

First, A Little History

Although the CEM 3310 was the first member of the Curtis line (introduced fall of 1978), it was actually preceeded by literally years of development and innovation which had resulted in many earlier designs, both discrete and integrated. The 3310 was the first commercially available IC envelope generator to offer extremely low control feedthrough (typically less than 10uV!) while other units were exhibiting 100s of millivolts of feedthrough, and often volts. The 3310 also offered precise and undistorted RC shaped contours, very exponentially accurate control scales, and tight tracking between units without trimming. Although the circuit technique developed by CES to achieve this performance eventually was used in competitive devices, the 3310 still remains unsurpassed in these features.

Writing For Chips Beats Singing For Your Dinner

To help motivate some of you aspiring Synthesource contributors to send your material in, we have decided to offer the following: For each circuit idea we publish in an issue of Synthesource, we will give the contributor his or her choice of any four chips we manufacture, absolutely free of charge. Depending on the chips selected, this offer amounts to a $32-40 value. For any feature article we use, two or more typewritten pages long, the contributor can select any 12 chips, amounting to a value of $96-120, or one of our CEV 3301 evaluation boards, value $89.95.

The circuit ideas or feature articles can be anything that would be of interest to our readers; that is, they do not have to use or be about our chips. However, to be perfectly honest, material concerning our products will probably have a better chance of being published.

So, it's time to stop your singing and start writing.

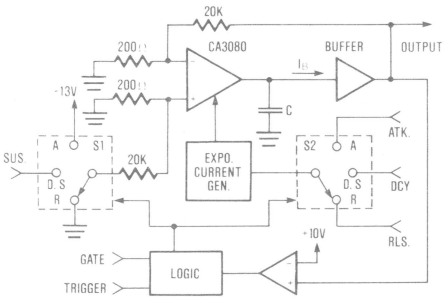


Figure 1: ADSR Using CA3080

The Traditional Method

Before the 3310, the most known and widely used method of getting a voltage controlled ADSR was to use the RCA 3080 in a configuration similar to that shown in Figure 1. Whenever the voltage to the non-inverting input of the 3080 is changed, the output always moves to balance the 3080 by forcing the negative input through the feedback network to equal the positive input. The resistor attenuators keep the 3080 in its linear region; thus, as the minus input approaches the positive input, a smaller and smaller portion of the exponential generator's current is fed from the 3080 output to the capacitor, resulting in an RC contour. The overall time constant of each contour is, of course, determined by the value of generator current and capacitor.

The major problem with this approach is the input offset of the 3080, which appears at the output 100 times larger due to the resistor attenuators. In a steady state, the output can be several hundred millivolts different from the desired voltage. More importantly, as the exponential generator's current is varied to change the time constant, the input offset may change, causing hundreds of millivolts of feedthrough at the output.

Another source of feedthrough is the input bias current to the buffer, which must be supplied by the 3080 output. Thus, even in the steady state, the 3080 is slightly imbalanced, and becomes more imbalanced as the exponential generator's current is reduced towards that of the buffer current. Again, this increasing imbalance at the 3080 inputs is magnified 100 times at the output.

Finally, the transfer function of the 3080 is non-linear, causing a distorted RC contour shape.

The Oberheim ADSR

At Winter NAMM in 1976, CES was contracted by Oberheim Electronics to design an ADSR on a chip for their programmer module which would not have these problems. The resulting technique CES developed revolved around precision one quadrant multipliers. The basic concept is shown in Figure 2. This circuit is familiar to most as simply an exponential current generator; but it is also a one quadrant multiplier with transfer function, $I_o = -I_{IN} \cdot e^{-Vc/Vt}$. Suppose the capacitor is initially charged to some value. Current I_o will begin to discharge C, but as the output voltage is lowered, so does I_{IN} and consequently the discharge current, I_o. Finally, when the output reaches zero, I_{IN} becomes zero, and there is no more output current I_o to discharge the capacitor further. The result is an RC discharge shape, where the actual time constant is again determined by the antilog of V_C.

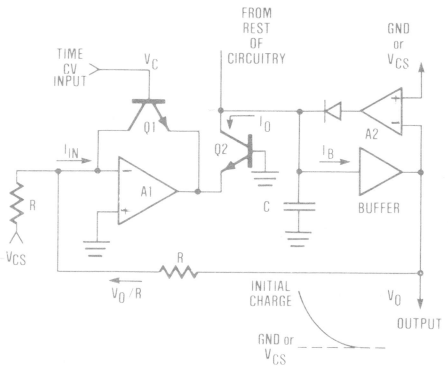


Figure 2: Basic Concept of Oberheim ADSR

The advantage to using a one quadrant multiplier is that as V_C is varied, there are no offsets which change and no currents to keep in balance, thereby eliminating one major source of feedthrough and output offset. The other major source, that due to buffer input current, is eliminated if the input current is in the direction as shown. The only problem remaining is that when I_o finally decreases to zero, I_b will continue to discharge C further. This problem is remedied with a clamping circuit made up of A2 which prevents the output from discharging more negative than the desired

steady state value.

This circuit as shown gives us the release portion of the envelope. The decay portion may also be realized by applying the negative of the sustain voltage, -Vcs, as shown. The output will then discharge with an RC contour only to Vcs, again becoming finally clamped by A2. In a similiar manner, the attack portion is realized with another one quadrant multiplier which operates only on the opposite polarity currents. In the final circuit developed for Oberheim, these one quadrant multipliers were switched in during the proper envelope phase to produce a complete ADSR envelope with absolutely no feedthrough, precise RC shape, and very accurate control scales.

The Birth of the CEM 3310

Some time after production was well under way (as will always be the case), a simpler method of implementing this concept occurred to the people at CES. Instead of switching one quadrant multipliers in and out, why not have a positive quadrant multiplier and negative quadrant multiplier peacefully coexist together, such that only one is conducting at a time. The result is a kind of class C two quadrant multiplier shown in Figure 3. The outputs O_1 and O_2 are complementary. Thus, when I_{IN} is positive, Q3 and Q4 are conducting, and O_2 is at $-V_{BE}$, causing O_1 to be at $+V_{BE}$, which turns Q1 and Q2 completely off. When I_{IN} is a negative current, Q1 and Q2 will now be conducting while Q3 and Q4 are cut off. Only one side is therefore conducting at a time, with the same benefits as the original Oberheim ADSR.

The other feature of this circuit is that the voltage setting the various steady state levels (asymptotes) can be applied directly to the plus input of the op amp. This is because the negative feedback will force the inverting input to exactly equal whatever voltage is on the non-inverting input. Suppose the output is at zero and V_A jumps to +13 volts; the inverting input will then also switch to +13, and I_{IN} will be negative (flowing from op amp input to output), causing Q2 to charge Cx at a rate determined by V_{CA}. The contour is RC because I_{IN} becomes less and less as the output rises.

Then, if V_A is switched to the sustain level, I_{IN} will become positive (flows from output to op amp minus input) and Q4 will begin to discharge Cx with an RC contour towards this sustain voltage and at a rate now determined by $V_{CD,R}$. When V_A is brought to zero, the output will then decay further until it reaches zero volts; the rate is again determined by $V_{CD,R}$, which may be changed during this phase to give a different time constant.

One final feature of this circuit configuration is that it also can be made to eliminate any feedthrough caused by the buffer input bias current, I_{b2}. With I_{b2} in the direction shown, it will continue to discharge Cx during the sustain and release steady state conditions. However, as soon as I_{IN} becomes slightly negative, Q2 will conduct enough to balance I_{b2} and

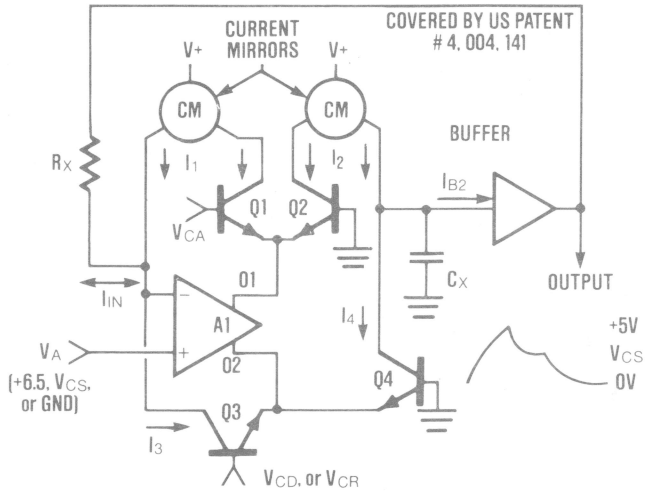


Figure 3: Basic Concept of CEM 3310 ADSR

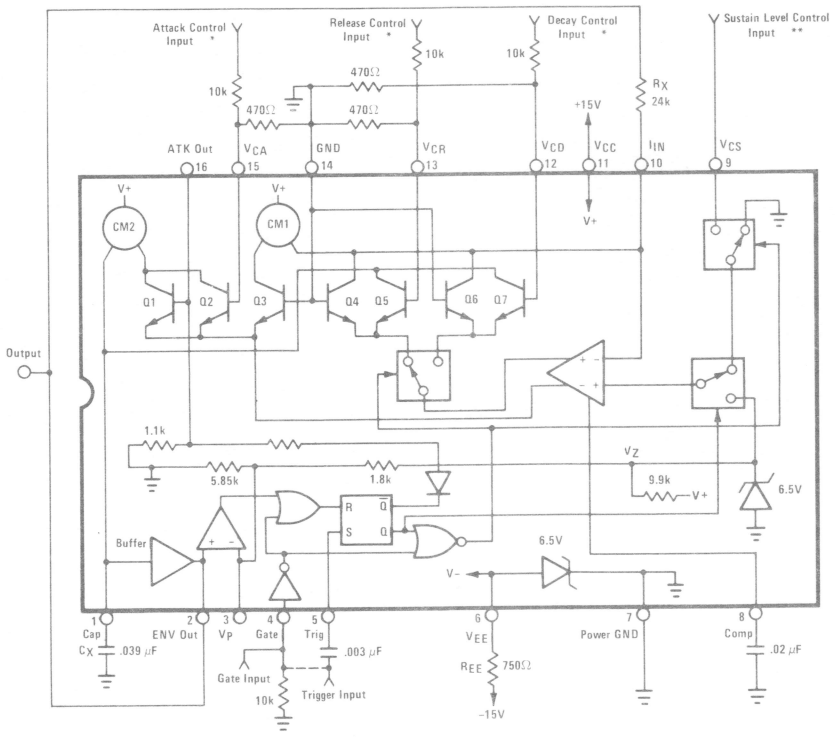


Figure 4: Block Diagram of CEM 3310

prevent Cx from discharging further. The problem is that as V_{CA} is varied, the I_{IN} through Q1 required to produce I_{b2} through Q2 will change also, causing the final resting point of the output to shift. Elimination of this attack control voltage feedthrough is simple, though: A circuit is added to connect the base of Q1 to zero volts during the decay-sustain and release phases, thereby preventing V_{CA} from having any effect and allowing the additional discharge amount to be only $I_{b2} \cdot Rx$, a very small value.

Although this circuit simplification had been devised, it was not put to use for some time. Finally, the clamoring from the industry for a low feedthrough, precision ADSR grew loud enough that CES added the additional elements shown in figure 4. The CEM 3310 was born.

Last Chance

We are giving all you slow pokes one more chance to subscribe to Synthesource before we drop you from the mailing list. If you have already subscribed, you have nothing to worry about, and we thank you very much for your support. If you have not yet subscribed, this will be your last issue until we see the dollar signs of your check or Money Order. Now, can a paltry sum of $12.50 per year be worth losing touch with the latest in technology and industry happenings?

HOT TIPS & CHIP BITS

The hot tip for this issue relates to computer tuning of oscillators. As most of you know, some of the commercial polyphonic synthesizers use an internal microprocessor to tune the initial frequency of all VCOs to the same note. Some also use the computer to adjust the control voltage scale factor as well. Adjusting the initial frequency can be as simple as just adding another Sample and Hold which supplies the frequency control input over a more limited range, and therefore with higher resolution, than the main S&H. Adjusting the scale factor, however, can be more difficult, often requiring a higher resolution DAC and more involved software.

The CEM 3340 VCO again shows its versatility. One big advantage to temperature compensating the oscillator with an on-chip precision multiplier is that this same multiplier can also be used to adjust the scale factor with an external voltage. Figure 1 shows two methods for adjusting the scale factor with the output of an additional Sample and Hold.

Manually adjusting the scale factor requires the resistance between pin 1 and V_{EE} (pin 3) to be adjusted with a trim pot. But since pin 1 is at a stable 3.01 volts above V_{EE}, the only purpose of this trimmer is to adjust the current flowing out of

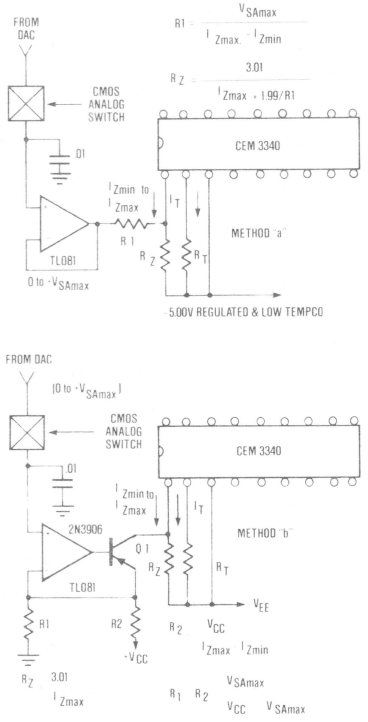


Figure 1: Methods for Computer Scale Adjust

pin 1 ±25% around the current flowing out of pin 2 (see data sheet or "Theory and Practice" of last issue). Therefore, this same task can be accomplished by feeding an adjustable external current into pin 1, which is precisely what these two circuits do.

Figure 1 also gives the necessary design equations using the minimum and maximum values of pin 1 current, I_{Zmin} and I_{Zmax} respectively, and the maximum value, V_{SAmax}, of the Sample and Hold output (assuming the minimum value is zero volts). The values chosen for I_{Zmin} and I_{Zmax} should be at least .75 and 1.25 respectively of the value chosen for I_T, the current out of pin 2. The method of 1a requires that the negative supply be -5.0 volts and that this supply be very temperature stable (better than 500ppm/^{O}C). The advantage of the method of 1b is that the negative supply is not critical and even the internal zener diode may be used.

The software necessary for these methods is fairly straightforward. First the scale adjust S&H is set to its mid-point and the frequency of the VCO set to around 100Hz. Then, the control voltage to the oscillator is switched between two values, corresponding to a desired two octave interval, and the oscillator frequency measured each time using the same hardware and software that is used for the initial frequency tune. The scale adjust S&H voltage is incremented in the proper direction each time through, most probably in a successive approximation manner, until the two frequencies are exactly two octaves apart. After this procedure, the initial frequency of the VCO is then tuned.

The only remaining parameter to be manually trimmed is the high frequency track, which is much less critical and easier to adjust than the first two. In fact, one can probably get away with a fixed resistor network setting this trim voltage at its nominal value.

Thus the CEM 3340 is an ideal candidate for complete computer adjustment. We hope that, armed with this tip, you will find the thought of computer tuning your system less formidable.

APPLICATIONS CORNER

Applications of the CEM 3345

Very often we receive questions about the CEM 3345 VCO, which is hinted at in our 3340 data sheet but not really explained. In our applications column this issue, we will attempt to shead some more light on the 3345 and offer a few specific applications, the most interesting of which is probably the circuit for linear FM through zero.

The CEM 3345 is exactly the same device as the CEM 3340 VCO, but with two more internal points bonded out onto an 18 pin DIP package. One of these additional points is the base of Q2, the transistor which performs the exponential function (see the data sheet). Internally grounded in the 3340, this base appears at pin 14 on the 3345, and in normal operation is simply connected to the external ground. Quite frankly, we have found no use for it, other than as an additional exponential control input (not temperature- compensated). We welcome anyone's ideas.

The other connection brought out is a buffered point of the switching comparator which appears at pin 7. This output is an open NPN emitter which switches nominally between a voltage equal to that at the hard sync input (-2.5V), and 2 diode drops below this voltage; thus when an external NPN is

connected as shown on the data sheet, the transistor will provide a nearly 50% duty cycle square wave, switching between V_{CC} and ground in step to the triangle wave.

Since most applications of our VCO require only the variable duty cycle pulse output, and since the 3345 is slightly more expensive than the 3340, virtually no manufacturer is in production with the 3345. The reason we offer the 3345 is that we thought an output signal from the switching comparator might be useful for something, and we actually had in mind a phase reversing VCO to produce, or at least simulate, the effects of linear FM through zero synthesis.

Linear FM through Zero

The circuit for a phase reversing VCO is shown in Figura 1a. The idea as presented by Bernie Hutchins in Electronotes[1] is that a negative frequency is the same as just a 180° phase reversal. Therefore, producing negative frequencies when the modulating signal swings negative simply requires that the instaneous frequency be determined by only the magnitude of the negative voltage (i.e. its absolute value), and that the direction, or phase, of the output waveform be reversed at the positive to negative transistion of the linear control input.

Direction reversal of the triangle output is precisely what the 3340 or 3345 is

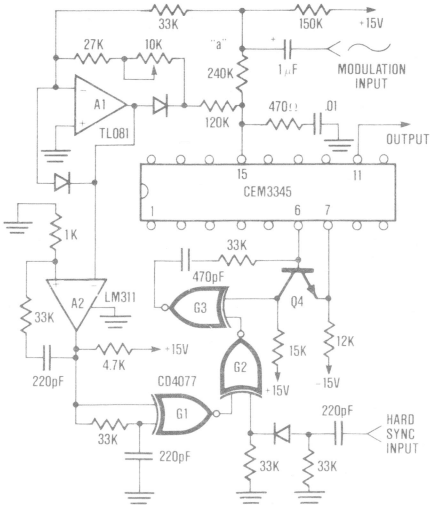


Figure 1a: Circuit for Linear FM through Zero

capable of doing with pulses applied to the hard sync input (pin 6). When the triangle output is rising, a positive pulse will switch the internal comparator and cause the triangle to ramp downward;& when the triangle is ramping downward, a negative pulse will switch the comparator into the other state, causing the triangle to now ramp upwards. The only trick then is to cause the triangle to always change direction whenever any pulse appears, and this is the purpose of the comparator output signal.

In the circuit of Figure 1, op amp A1 takes the absolute value of the FM modulation input as it appears at point "a", the resistor-capacitor junction. With the resistor values as given, this point sits at approximately +2.5V with no signal, providing +10uA of reference current into pin 15, the linear control input. Thus a 5VPP signal can be accomodated before the input is theoretically swept through zero.

For modulating signals greater than 5VPP, the output of A1 swings through zero volts, switching comparator A2 every time point "a" passes through zero. Exclusive NOR gate G1 converts every output transition of A2, whether positive or negative, into narrow negative pulses which are inverted by gate G2. Gate G3 then gives these pulses the proper polarity for reversing the direction of the triangle wave: When the triangle is rising, the collector of Q4 is high, making G3 non-inverting; when the triangle is falling, the Q4 output is low, and G3 is inverting.

Needless to say, the sounds produced by this circuit are quite dramatic. Theoretically, the apparent pitch of the output should be that of the modulating wave (we'll restrict our discussion to sine waves as the modulation source); as we increase the modulation depth until negative frequencies are just being produced, we find that the timbrel transistion is smooth and the apparent pitch remains the same, attesting to the fact that perhaps we are indeed FMing through zero. Figure 1b shows a typical waveform produced with the carrier frequency at 1KHz, the

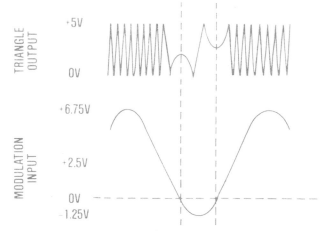


Figure 1b: FM through 0 Waveforms

modulating frequency at 100 Hz, and the modulation index at approximately 15. As can be seen, the triangle does reverse directions at the zero crossings.

The one major problem we have observed is internal beatings of the output as heard and seen on an oscilloscope.Theoretically, if the carrier frequency is an even multiple of the modulating frequency, then the new sidebands created will be harmonic. However, we observed that as we increased the modulation depth, these beatings would begin to occur, especially after we passed through zero, indicating that the sidebands being produced were becoming non-harmonic. As the depth was increased further, the sound would go through alternate states of beating very rapidly and not beating at all. The trimmer with A1, whose purpose is to trim the absolute value circuit, did not seem to help. Any suggestions from our readers would be very much appreciated.

Yet Another Hard Sync

The circuit of figure 1 also includes an external input labeled "hard sync". With the hard sync methods produced by coupling pulses directly into the hard sync pin (as described in the data sheet), the triangle waveform is not necessarily affected, or synchronized, by every pulse. However, gates G2 and G3 of figure 1 allow every positive transition at the "hard sync" input to reverse the direction of the triangle wave, and thereby synchronize it. The resulting sound not surprisingly is somewhere between the other hard sync methods and the sound of linear FM through zero. If this is the only effect desired, then of course A1, A2, and G1 can be eliminated, and we have our second application for the square wave output of the 3345.

Adjustable Triangle Symmetry

The final application for this output came in response to a request from a customer who needed to generate variable triangle symmetry. Figure 2a shows the circuit. Everytime the triangle reverses directions, gates G1 and G2 switch states, feeding either more or less reference current, depending on

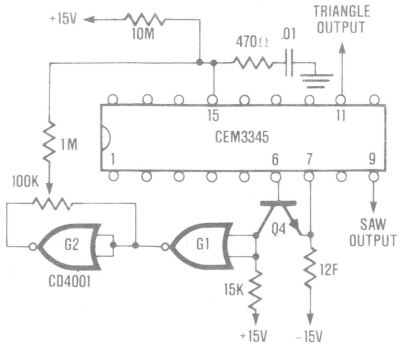


Figure 2a: Circuit for Variable Symmetry

the position of the pot, into the linear control input pin which in turn changes the slope rate. With the values as shown, the slope ratio can be varied between 10:1 and 1:10. The effect this has on the sawtooth output is also quite interesting: Figure 2b shows the range of waveforms possible. The

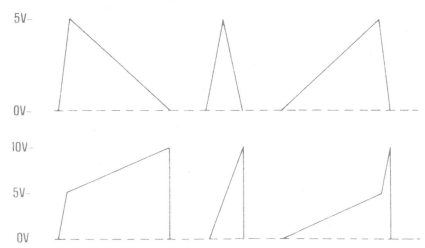


Figure 2b: Variable Symmetry Waveforms

one disadvantage of this circuit is that the frequency will also change as the symmetry is varied. For best stability, CMOS logic should be used for the gates.

For Those With Only a 3340

In concluding, we should mention that at least the last two applications can

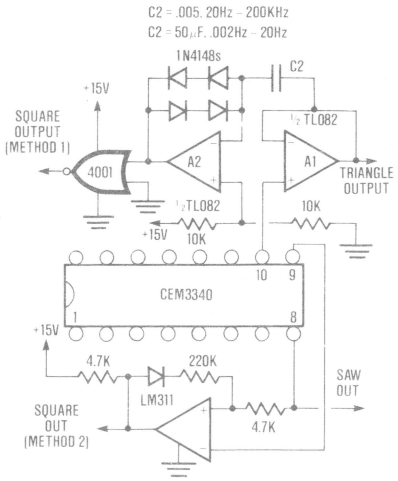


Fig 3: Deriving Square Wave from CEM 3340

be implemented using the CEM 3340 with some additional circuitry. Figure 3 shows two methods of deriving a square wave which corresponds to the triangle wave. In the first, the rising or falling slopes of the triangle wave are differentiated into positive or negative currents by A2. Using diodes for feedback

elements keeps the output swing of A2 relatively constant over frequency. These diodes may be substituted with two 2.7 volt to 6.8 volt zener diodes back to back if desired. The triangle output should be buffered, as with A1, because the currents through C2 can be greater than 1mA at the higher frequencies. with this circuit, there is a small delay of several microseconds between the slope change of the triangle waveform and the square wave transistion.

The second method shown in Figure 3 simply compares the sawtooth output with the internal comparator reference voltage on pin 9. Whenever the triangle reaches its peak, the sawtooth output is always within ±25mV of the reference voltage; thus the comparator A3 output will always switch states within 0.5% of the triangle half period.

The first method can not be used with linear FM through zero because the slope of the output waveform is changing and often flat. With the second method, the square wave may not be coincident enough with the triangle wave. For FM through zero, then, the 3345 will be the best choice. But the 3340 can probably suffice in most other applications.

1. B. Hutchins, "New Design Ideas for Voltage Controlled Oscillators," Electronotes, Vol. 8, No.62 Feb., 1976.
"Time Reversing Oscillators and Through-Zero FM," Electronotes, Vol. 12, No. 115, July, 1980.

EUROPEAN SCENE

Perhaps unknown to many of our fellow Americans, synthesizer activity in Europe is quite lively. In northern Europe (England, Holland, France, Germany, Scandinavia), this activity seems to be concentrated more in the home constructors market than in actual manufacturing of commercial instruments. Although there are a number of small companies in these countries producing finished synthesizers, what seems to be more abundant is synthesizer kits of all types, sizes, and shapes as well as numerous articles in hobbyists magazines to back them up. These kits range from a P.C. board and a bag of IC's that one can pick-up at any one of numerous hobby electronics stores to a complete kit with all parts, finished case, and a step by step assembly manual, much like Heathkit. And almost every issue of a popular electronics magazine, such as Electronics Today International and Elektor has a feature on synthesizer circuits. There are even several totally devoted to the subject--Electronics & Music Maker from England and Musik-Elektronik from Germany.

As one might expect, the Curtis line of chips has become part of all this fervor. Used in many of these kits and home constructors projects, they have become quite well known, mostly due to the efforts and technically excellent articles written by our English and Dutch distributors in these magazines. Hopefully, we will reprint some of these articles in future issues of Synthesource.

In southern Europe, specifically Italy, the story is quite different. Beginning with their famous accordians, Italy has always been known as a world producer of musical instruments. With the advent of electronics, they have become the largest producer and exporter of home electronic organs. We have always been impressed with the expertise and creativity of their engineering and use of the latest technological advances. Some very interesting synthesizers are now beginning to emerge from Italy.

A listing of our European distributors is provided below. Not only do they stock our products, but many sell kits and/or evaluation boards using our devices, and all will provide expert applications assistance, if required.

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-Interview with a prominent Bay Area Studio Musician (to get the other point of view on commercial synthesizers)

-First Part of a Series: Review of VCA Techniques

-Voltage Controlled Transconductors in non-VCA Applications

-And more Readers Circuits, More Hot Tips and Chip Bits, More European Scene, and just More, More, More!

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